(54) Title: INTERNET TELECOMMUNICATIONS SYSTEM

(57) Abstract

A method and apparatus is described which enables users (10) of a standard telephone system (12) to access a packet-based network, such as the Internet (36). A first callover unit (32) converts signals (14) from the standard telephone system (12) into packets (34) formatted for Internet transmission. Embodiments are described which allow a voice telephone call to be placed substituting the Internet (30) for a long-distance carrier (22). A second callover unit (40) converts the packets (38) back into voice signals (24) transmittable over a standard telephone network (20) to a standard voice telephone (28).

Title: Internet Telecommunications System

Background of the Invention

This invention relates to voice and data telecommunications and, in particular, to a telecommunications system which interfaces between a standard telephone and the Internet to provide reduced-cost long-distance service over the Internet.

Conventional telephone systems such as that depicted in Figure 1 have evolved over time
5   to provide a reliable means of communication between two points. In Figure 1, the two points are standard telephones used for voice communication but modems can easily be substituted to allow computers to communicate.

The standard telephone system requires a first telephone 10 to initiate a call into a local exchange company ("LEC") 12 which provides regional phone service. The first telephone
10   10 is defined as a conventional telephone designed for use with telephone carrier networks, such as a T-carrier network which uses T1-based digital transmission methods, for example. Such a system will hereinafter be referred to as plain old telephone service ("POTS").

For a long-distance call, the LEC 12 then connects to a regional switch 16 of a long-distance carrier 22 and transmits a voice signal 14. The voice signal in this instance is
15   directed to a destination by a numeric code, usually an American numbering index ("ANI") code. The regional switch 16 transmits the voice signal 14 via the long-distance carrier's trunk 18 to a second regional switch 20 for the destination region. A voice signal 24 is then transmitted to a LEC 26 for this region which connects to the second telephone 28. A problem with such a system is the cost to use. With each of the LEC's 12, 26 and the long-
20   distance carrier each charging for the call, usually charged at a rate per unit time which has to support the costs of their individual networks, a long-distance call can become very expensive.

A further problem arises when one of the telephones 10, 28 is located in a country which has an inferior telephonic infrastructure. In this case, each caller must vie for limited

telephone lines, often to no avail. The infrastructure necessary to enhance the long-distance service is also costly making rapid enhancement of the telephone network unlikely.

In contrast, computer networks have grown rapidly throughout the world. Information dissemination and retrieval via computers has become a mainstay of commerce. The Internet

5   has formed to unite various computer networks into a more cohesive integrated system. The Internet has developed into a world-wide system of computers performing a function of content servers where the content servers are owned and maintained by governments, educational institutions and commercial entities. Since these content servers are independently operated and maintained, the cost to use the Internet is usually only a nominal

10 monthly fee required for the Internet's main transmission lines, or backbone. A limitation of the Internet is, therefore, that it is only accessible by computers.

Speed of transmission over the Internet can at times be a problem. Though current Internet technology can accommodate connections to approximately twenty times the number of computers in existence today, the backbone of the Internet can suffer from traffic

15 congestion, commonly referred to as a bottleneck. The bottleneck limits throughput due to competition for IP paths. This means that as more computers come on-line, transmissions will slow due to increased transmission errors resulting in retries. And since each computer is allocated only one address, redundant communications are virtually impossible. Further, packet size is usually matched to the maximum transfer rate of the network such that the least

20 overhead can accompany the greatest quantity of data. The Internet though, is a network of networks so knowing the maximum transfer rate is virtually impossible since a sender never knows the path that the transmission will ultimately take. If the packet is routed through a router having only a small bandwidth then the router automatically breaks the packet into multiple packets before passes them through. This process delays the overall transmission

25 speed thereby affecting transmission quality for real-time transmissions.

Several commercially available products have recently emerged that enable voice communication over the Internet. Each of them use a sound card in a personal computer ("PC") to translate between voice and electronic signals. The electronic signals are then packetized by an internet engine provided by an Internet subscription service. The receiving

end of the transmission must also have a PC with a sound card and a subscription to an Internet service. Along with the obvious hardware and subscription constraints to this system, the voice transmission quality is unacceptable. Having access to only one Internet address as previously mentioned, voice transmissions are subject to errors, and thus retries.

5 Further, an additional protocol, usually point-to-point ("PPP") or serial in-line protocol ("SLIP"), must be added for the transmission from modem to modem thereby increasing overhead. In practice, this results is delays and spaces distributed throughout the speech, including in the middle of words and sentences, that can be heard on the receiving end.

Therefore, there is a need to integrate POTS communications into the Internet such that
10 the Internet can be used as a long-distance service with acceptable quality and a POTS interface.

Accordingly, it is an object of this invention to reduce a cost of long-distance service.

It is another object of this invention to use an existing computer network infrastructure to enhance long-distance services globally.

15 It is still another object of this invention to allow POTS communication to content servers located on the Internet.

It is a further object of this invention to increase transmission speed, transmission quality, and reliability over the Internet.

It is a further object of this invention to provide access to the Internet for both voice and
20 data communications without a need for a user to buy computer hardware or to have a subscription to the Internet.

These and other objects of the invention will be obvious and appear hereinafter.

Summary of the Invention

The aforementioned and other objects of the invention are achieved by the invention which provides a telecommunications system for bi-directionally communicating information signals between a first communication device which is electrically connected to a telephone network and a second communication device. In one embodiment, the telecommunications system forms a long distance telephone system by connecting the first communication device to a first interface means through the telephone network. By way of example, the first communication device can be a telephone communicating to the interface means through a local telephone exchange.

To use the long distance telephone system, a call is placed to the first interface means. The first interface means then adaptively determines a best-quality, low-cost method to complete the telephone call. The methods available comprise traditional long distance service, or various transmission methods of transmission over a network means. Performance of the traditional long distance carrier is substantially constant and known. Therefore, a comparative determination is made by periodically testing the network means over which the call will be sent.

The network means is a network of a plurality of computers or computer networks linked together electronically. Communication is performed using discrete packets of digital information which are addressed to a specific computer or communication device. An example of such a network is the Internet.

To achieve high-quality communication for real-time transmissions, such as voice, the interface means transmits over multiple IP streams redundantly. Such transmissions increase the possibilities that the transmission will be received in a single try with no gaps.

A second interface means also connected to the interface means receives the discrete packets and assembles them such that they can be retransmitted over a second telephone

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network to another telephone. Since the first and second interface means operate in two modes, transmitting mode and receiving mode, bi-directional communication is achieved.

In transmitting mode, the interface means converts information signals from the exchange means into discrete packets of digital information and transmits the discrete packets

5 to the network means. In the receiving mode, the interface means converts the discrete packets received from the network means into information signals and transmits the information signals to the exchange means. This continues bi-directionally until the call is complete.

In a second embodiment, the telecommunication system is being used to interact with the

10 network means directly. For example, a computer having a modem but not normally having Internet access can access the Internet; a POTS telephone can access a content server on the Internet to leave voice mail or interact with a menuing system. To accomplish these tasks, the first communication device contacts the first interface means, as before, over the telephone network, POTS for example.

15 The interface then converts information sent from the first communication device to discrete packets and sends them over the network means. If the second communication device is a content server on the network means then it responds with discrete packets of its own. The interface means then translates the discrete packets of the content server into information for the first communication device thus establishing bi-directional

20 communication.

In further aspects, the invention provides methods in accordance with the apparatus described above. The aforementioned and other aspects of the invention are evident in the drawings and in the description that follows.

Brief Description of the Drawings

The foregoing and other objects of this invention, the various features thereof, as well as the invention itself, can be more fully understood from the following description, when read together with the accompanying drawings in which:

5   Figure 1 is a block diagram of a prior art long-distance communication system;

Figure 2 is a block diagram of long-distance communication system in accordance with the invention;

Figure 3 is a block diagram of a callover unit which is a part of the long-distance communication system shown in Figure 2;

10   Figure 4 is a flow chart the callover unit of Figure 3;

Figure 5 is a block diagram of a billing and management server user with the long-distance communication system of Figure 2;

Figure 6 is a block diagram of a computer communicating with a POTS telephone and a computer over the long-distance communication system of Figure 2;

15   Figure 7 is a block diagram of a computer communicating to another computer over the Internet where the first computer does not normally have Internet access;

Figure 8 is a graphical user interface for software required in the computer shown in Figure 7; and

Figure 9 is a block diagram of a standard telephone communicating to a computer over
20   the Internet using the long-distance communication system of Figure 2.

Detailed Description

The invention achieves the aforementioned goals by replacing the long-distance carrier 22 shown in Figure 1 with an Internet communication subsystem 30 as illustrated in Figure 2. Figure 2 shows the telephone 10 calling the LEC 12 through which the voice signal 14 is
5　routed. One skilled in the art will realize that the telephone 10 is generically referring to any of various telecommunications devices which communicate over telephone lines. For example, the telephone can be a common type-500 telephone for voice communication, a facsimile machine modem or other such modem, video phone, pay phone, pager, cellular phone, et cetera.

10　The voice signal 14 is then transmitted by a callover unit 32 instead of a long-distance carrier 22. The callover unit 32 converts the voice signal 14 into packets 34 which are time-division multiplexed and transmitted over the Internet 36. Once routed through the Internet 36 to a proper region, the packets 38 are reassembled in a second callover unit 40. The second callover unit 40 reassembles the packets into a voice signal 24 and transmits the voice
15　signal 24 through the LEC 20 to a second telephone 28. As previously stated, the second telephone 28 can be any of various telephonic devices commonly operating over telephones line and should not be restricted to common voice telephones.

It should be noted that when the telephone 10 initiates the call, the LEC 12 provides a dial tone to the telephone as is commonly known in the art. And, if the call where placed
20　within the geographical area serviced by the LEC 12 then feedback such as ringing, busy signals, et cetera, would also be provided by the LEC 12. Using the traditional long distance service as is described in Figure 1, these indications would be provided by the destination LEC 20 and automatically passed back through the long distance carrier 22. In the invention though, this feedback must specifically handle this call setup, call progress and other POTS
25　signaling. Therefore, the feedback is sent back from the destination LEC 20 to the second callover unit where it is packetized and transmitted over the Internet 36 to the first callover unit 32. The first callover unit 32 then reassembles the packets to form a signal transmittable

over LEC 12 to the telephone 10. From a users standpoint, the signals sound substantially identical to those transmitted over the traditional long distance service.

Though only one signal direction is described, one skilled in the art will realize that the system is bi-directional and that signals returning travel a substantially identical path to that
5   of the first direction.

Additionally, billing can be accomplished through the Internet by attaching a billing unit 42. The billing unit 42 is a server on the Internet which is addressable by the callover units 32, 40 to ensure accurate billing.

In the practice of one embodiment, the telephone call is originated from the telephone 10
10   which dials a local phone number within the region of the LEC 12 to contact the callover unit 32. The callover unit 32 answers the phone and provides a menu from which the user chooses. Choices include, but are not limited to, Internet service or service by various long-distance carriers. Henceforth, unless otherwise stated, it will be assumed that Internet service is selected.

15   Upon being prompted, the destination phone number is keyed via the keypad on the telephone 10. The callover unit 32 then queries an internal database to determine a remote callover unit 20 that located geographically close to the destination telephone 28. Preferably, this is in the region of the LEC 20. If it is not in the region of the LEC 20, a long-distance carrier is used to bridge the distance in between.

20   To understand how the invention reduces costs as well as to completely understand the invention itself, IP networks such as the Internet must first be understood. As previously described, the Internet is essentially a network of sub-networks. In order for each of the sub-networks to communicate with other sub-networks they must use a set of predetermined protocols; the most basic two of which are the Internet Protocol (IP) and the transmission
25   control protocol (TCP). These two protocols along with other supporting protocols are known as the TCP/IP Internet Protocol Suite ("TCP/IP"). Utilizing TCP/IP provides a

foundation to the Internet which is based upon the transmission of data packets. Data packets are also generally essential to costs saving in data communications. Communicating with data packets ensures that individual wires need not be dedicated to each pair of communicating computer but, instead, multiple computer share hardware facilities by

5   packetizing. Though what each packet actually contains is network specific, each packet commonly contains addressing information which ultimately determines which computer will receive the packets. Since every computer on the Internet has a unique address, called the IP Address, each packet is then properly routed through the Internet. Further, since TCP/IP is an ordered protocol, packets are received on a first in first out ("FIFO") basis so that a

10   transmitting device does not have to mark each packet in the order that it was sent for proper reassembly.

With that basic background, Figure 3 can be examined. Figure 3 shows a callover unit 32 broken out functionally. As the functionality is described, example hardware implementations are stated. These hardware implementations should be considered

15   illustrative of the preferred embodiment and are not in any way restrictive.

The voice signal 14 arrives in the callover unit 32 at the line interface 50. The line interface is a preferably a multi-port digital card, such as a T1 card manufactured by Digilogic, Inc., but could also be an analog card having analog-to-digital conversion, an ISDN card, or other card for conforming with any of various standards in telephony.

20   Regardless of the input format, the output of the line interface 50 is transmitted to a matrix switch 52 which routes the voice signal 14 internally to the IVR 54. The matrix switch 52 in the preferred embodiment is a time-division multiplexed bus which is incorporated into the IVR card described below.

One skilled in the art will realize that additional calling features such as voice and data
25   bridging can also be incorporated into the callover unit for multiple-party calls.

The IVR 54 is the previously mentioned interactive voice response ("IVR") module that presents menu options to the user. In the preferred embodiment, dual-tone multi-frequency ("DTFM") signaling is used for touch-tone telephone systems. An analogous system can also be used for pulse dialing systems. Along with the aforementioned choices of long-distance

5   communication method, the IVR 54 presents options for credit card or third party billing, et cetera. Billing choices are passed to the billing manager 66 which is later herein described.

Assuming that Internet service is chosen, the voice signal 14 is passed to a digitizer 56 which digitizes and compresses the voice signal 14. If the voice signal is already digital then the digitizer 56 simply compresses the voice signal 14 using lossless compression techniques

10 well known in the art. In terms of hardware implementation, the digitizer 56, the IVR module 54 and the matrix switch 52, in the preferred embodiment, are all on a single IVR card such as the D-24CC-T1 card manufactured by Digilogic, Inc.

Packetization is then performed by the packetizer 58. Packetizing is performed such that the resulting packet conforms to the Internet Protocol (IP). The packets are also scaled in size

15 to ensure expedient transmission. Instead of matching the packet size to a maximum transfer rate of a known part of the system as was previously described, the packetizer matches the packet size to a maximum transfer rate of the transmission path. A scaled packet size is then used which virtually ensures transmission without parsing through substantially all routers. The scaled packet size is determined during quality testing which is later herein described.

20 Each packet incorporates a header which includes a destination IP address, the sending IP address as well as other administrative information. The sending IP address is simply the IP address of the callover unit 32 which is sending the packets. The destination IP address is determined as follows.

The destination phone number keyed during the menuing step is used to query an internal

25 database to determine a remote callover unit 20 that located geographically closest to the destination telephone 28. The area code in the phone number is first used to determine a

group of callover units servicing that area code. The exchange then determines which of those callover units is a proper choice for the destination telephone 28. If multiple callover units are in the region of the LEC 26 for the destination telephone 28, then the originating callover unit 32 determines which to use by testing to see which will take the least time.

5   Further, since connection time is considered an important measurement of quality, the callover unit 32 tests and maintains predictively certain IP routes, thus speeding availability of many common destinations.

The internal database is relational in the preferred embodiment. The database relates each callover unit in on the Internet with area codes and exchanges. Further, each callover

10   unit has a list of IP addressees associated therewith. In the preferred embodiment there are two hundred fifty-five IP addresses per callover box which are stored in a table as an array of addresses. Since voice and data communications would garble if the same IP address and, thus IP stream, were used by different callers, the database tracks which IP streams are already being used and indicates which is the next available IP stream. An IP stream shall be

15   defined as a transmission path having a specific IP address which carries asynchronous data repeatedly.

The packets are then sent to an internet engine 60. The internet engine 60 is usually a software package that handles the transmission control protocol (TCP) as well as other administrative tasks. The internet engine 60 in this embodiment is a proprietary UNIX-based

20   package which performs such tasks as sending the packets and monitoring returning packets for an acknowledgment of receipt of the packets. If such a receipt is not received within a limited time period, three second for example, the packet is retransmitted. Another task performed in the internet engine 60, inter alia, is disregarding repeated packets that were erroneously retransmitted.

25   The internet engine 60 then transmits the packets to IP channelization 62. IP channelization determines how many virtual IP streams will be used for transmission. Using

a method later herein described, the packets are sent out over multiple IP streams through the Internet interface 64 which is a hardware connection to the Internet 36. Again, the Internet interface 64 is a T1 card, in the preferred embodiment.

Using multiple IP stream allows the callover unit 32 to both minimize average latency in data transmission while also maximizing reliability. Latency is the delay between transmission origination and transmission receipt. In the case of a voice telephone call, the latency is the delay between when someone speaks and when it is heard. The callover unit 32 decreases latency by breaking the voice transmission into pieces and nearly simultaneously transmitting the pieces over multiple IP streams. Reliability is increased by transmitting the same pieces multiple times, thus increasing the chance that the piece is received at the other callover unit 40 on a first transmission of that piece. In the previous analogy, a missed piece of the transmission would be heard as a skipped word or part of a sentence.

The following table illustrates both that lossless compression is used since all bytes of voice and data are at all time accountable, and how using multiple IP streams relate to packet size and thus, can decrease latency.

Real Time Data Chart for Voice Packets				
Voice channel rate:	64 Kb/s = 8000 Bytes/s		32 Kb/s = 4000 Bytes/s	
	100 ms = 800 Bytes		100 ms = 400 Bytes	
	Bytes/stream		Bytes/stream	
milliseconds	5 streams	2 streams	5 streams	2 streams
100	160	400	80	200
200	320	800	160	400
300	480	1200	240	600
400	640	1600	320	800
500	800	2000	400	1000

Figure 4 illustrates the decision making process undertaken in the callover unit 32 and how latency is minimized. Once the process is started 80, i.e., a user calls into the callover

unit 32, the user preferences are determined 82. The most basic of these preferences is whether to use a long-distance carrier 84 or the Internet 86. If the long-distance carrier is chosen 84 then the call is connected through a choice of long-distance vendors 110.

5 If the Internet is chosen, then the next question is whether the transmission is latent sensitive 88? In other words, must the transmission be performed in real time? The answer 'NO' 90 indicates that the transmission is not latent sensitive. Examples of such transmissions are facsimiles, computer-to-computer, et cetera, i.e., purely data transmissions.

A latent sensitive transmission is, for example, voice. Two people trying to communicate by phone would find delays in transmission unacceptable. Therefore, if it is 10 determined that the transmission is latent sensitive 92, then latency minimization is performed.

First the callover unit 32 determines whether there are sufficient IP streams available 94. The actual number of IP streams is implementation specific. In the preferred embodiment, the minimum number is two, below which the callover unit will not continue 96 and the call 15 is shifted to a traditional long-distance carrier service 110.

If there are sufficient IP streams 98, a test is performed to determine the average latency in transmission. The average latency, or the average transmission delay, T_{delay}, is defined algebraically as

$$T_{delay} = T_c + T_a + T_s + T_d + T_p$$

20

where

T_c - time to collect voice data

T_a - time to assemble into packets

T_s - time to send the data over the Internet

T_d - time to disassemble

25

T_p - time to pass through digital-to-analog conversion

As can be seen from the equation, most of the latency variables are within the control of the callover unit. T_c, T_a, T_d, and T_p are each hardware dependent, in the preferred embodiment, and, therefore, if faster hardware is used then latency is reduced. T_s is dependent upon the Internet though. The Internet has a backbone which is a main line for

5   communication. When the backbone incurs traffic, latency T_s will increase. Therefore, the next step is to determine which transmission method will minimize latency given the current state of the Internet backbone.

Generally, latency T_s must be minimized but without sacrificing data reliability. The goal for reliability is near error free delivery with up to 500 millisecond voice delay in one

10   transmission direction. Using multiple IP streams for a single telephone transmission eliminates dependency on time-consuming packet retries, thus averaging transmission performance to a fixed quality level.

In the preferred embodiment, latency T_s is further minimized by defining separate IP streams for transmitting and receiving. This enhances performance because IP streams are

15   not isochronous and, therefore, unidirectional communication along an IP stream can be better measured and sustained by communicating unidirectionally. Additionally, small packet sizes can be used to allow faster reconstruction of the packets from multiple IP streams into a single cohesive voice signal.

To minimize latency T_c, a controlled buffer size is used to collect data and then the buffer

20   is transmitted using the least latent transmission method possible. This buffer size is scaled to the maximum transfer rate of the transmission path as was previously described.

To determine the best transmission method, various considerations must be evaluated. Empirically, the size of the buffer must change proportionally to the voice channel rate, i.e., packet size as exemplified in the previous table, but the latency T_c should not exceed a value

25   where it is more than twenty percent of the total transmission delay, T_{delay}. Should the transmission delay, T_s, become so low such that the buffered period in bytes of data is greater

14

than twenty percent of the transmission delay, then the number of IP streams must be increased and the buffered period will be sized to approximate the packet size.

Further, the number of IP streams for a call is generally determined by taking the total number of IP streams (minus those reserved for system use) divided by the sum of a number 5 of voice call requests plus the number of calls in progress or an average for that hour, whichever is greater.

The control of latency is, therefore, dynamic and adaptive to the current state of the Internet.

An example method is shown for determining latency on the Internet that is used in the 10 preferred embodiment. The example uses four passes, A-D, and five IP streams. One skilled in the art will realize that more or less passes and IP streams can be used and that these quantities are also adaptive as described above.

For each 100 millisecond period at 8000 bytes per second, test the streams for least latent passage.

Pass A	IP Stream 1 —	send ¼ of data four times
	IP Stream 2 —	send ½ of data twice
	IP Stream 3 —	send all the data once
	IP Stream 4 —	send ½ the data simultaneously with stream 5
	IP Stream 5 —	send ½ the data simultaneously with stream 4

15 For each 200 millisecond period at 8000 bytes per second, test the streams for least latent passage.

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Pass A	IP Stream 3 — send all the data once
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	IP Stream 5 — send ½ the data simultaneously with stream 4

15   For each 200 millisecond period at 8000 bytes per second, test the streams for least latent passage.

Pass B	IP Stream 1 — send ¼ of data four times
	IP Stream 2 — send ½ of data twice
	IP Stream 3 — send all the data once
	IP Stream 4 — send ½ the data simultaneously with stream 5
	IP Stream 5 — send ½ the data simultaneously with stream 4

For each 100 millisecond period at 4000 bytes per second, test the streams for least latent passage.

Pass C	IP Stream 1 — send ¼ of data four times
	IP Stream 2 — send ½ of data twice
	IP Stream 3 — send all the data once
	IP Stream 4 — send ½ the data simultaneously with stream 5
	IP Stream 5 — send ½ the data simultaneously with stream 4

For each 100 millisecond period at 8000 bytes per second, test the streams for least latent passage. This test redundantly transmits the same portion of the data simultaneously over multiple IP streams. Since this is remarkably redundant, retries on any of the IP streams, up to n-2 where n is the number of IP streams, cause a termination of that IP stream as unusable.

Pass D	IP Stream 1 — send 1/5 of data
	IP Stream 2 — send 1/5 of data
	IP Stream 3 — send 1/5 of data
	IP Stream 4 — send 1/5 of data
	IP Stream 5 — send 1/5 of data

This four-pass method is repeated periodically to update the transmission method incurring the least latency. Such testing is performed asynchronously with communications, including real-time communication such as voice, without noticeable interruption to the

quality of the communication. In the preferred embodiment, the test is repeated once every three seconds.

Once the best average latency is determined 100, the best mode of transmission is determined 102. If none of the four passes in the above example gives acceptable latency or
5   data error rates were too high then a best mode is deemed to have not been determined 104 and the transmission is shifted to a long-distance carrier 110.

If a best mode of transmission is determined 106, then the call is routed 108 through a callover unit in the geographical region of the destination telephone 28. Alternatively, if there is unacceptably high latency in only a segment of the transmission path, the best mode
10   may be through a callover unit outside of the geographical area and traditional long distance service is used to bypass the slow segment of the transmission path. This is described in detail later herein.

As the call continues, the Internet is repeatedly tested to minimize latency and data errors. If additional IP streams become available the callover unit 32 can increase the number
15   used to transmit the telephone call. Likewise, as demand on the callover unit 32 increases, i.e., more telephone calls are placed, IP streams can be withdrawn from current telephone calls to accommodate the increased load.

When the telephone call is completed 112, the resources are freed so that a subsequent caller can use them.

20   Figure 5 shows a billing and management system for tracking phone calls placed over the Internet. In this illustration, the LEC was replaced by a public switched telephone network ("PSTN") 120 which is generic for a telephone system. The PSTN 120 can be a LEC or can include long-distance carrier, or can be a foreign equivalent. In any of these cases, the PSTN 120 has associated therewith a third-party billing system 122.

25   The third-party billing system 122 is one of the many billing systems used by telephone companies to charge for telephone calls. Because of the billing capabilities in the invention,

the callover unit 32 can selectively interact with the third party billing system 122. Therefore, an Internet process such as the aforementioned long-distance service can be billed directly on regular telephone billing systems. For example, when a user places a call to the callover unit 32, one of the options in the menu can be to bill a third party in which case the

5   charges are sent back through the PSTN 120 to the third-party billing system 122.

Once a call is established by the callover unit 32 through the Internet 36 to a destination, charges begin to accumulate. The charges are temporarily stored within the callover unit 32 in an internal database 124. Once the call is terminated, the charges cease and the final amount stored in the temporary database 124 is sent over the Internet 36 to a billing and

10   management server 42 where they are stored in a larger database. The larger database is ultimately used to generate call detail recording ("CDR") invoices for the calls made over the Internet telecommunications system.

The billing and management server 42 also maintains the databases in the callover units. Each time a new callover unit is placed in service, or exchanges associated with each callover

15   unit change, for example, the billing and management server 42 updates the databases contained in each of the callover units. This is accomplished by downloading over the Internet an updated database containing IP Address information for the new callover unit, area code(s) and exchanges which the callover unit will service. The database is also encoded prior to transmission for security.

20   Another function performed by the billing and management server 42 is to update latency information in each of the callover units. Each of the databases in the callover units contains latency information for segments of the transmission path, i.e., the Internet. This latency information is stored in the individual callover units representing calls made over the previous twenty-four hour period, in the preferred embodiment. Periodically, the billing and

25   management server 42 uploads the latency values from each of the callover units to update its own database. Each time a segment of the transmission path deviates from its stored latency

values by a predetermined quantity, the billing and management server 42 downloads the new latency values to the individual callover units.

Having latency values allows the callover units to bypass known problems. For example, over the past day the transmission over the Internet between London and Paris have been
5 extremely slow. A caller in Boston attempts to call someone in Paris. Since the callover unit in the Boston region knows about the latency problem, the call can be sent to the callover unit in London over the Internet instead of the callover unit in Paris. The callover unit in London can then access a traditional long distance carrier for communication between London and Paris, thus avoiding a problematic delay.

10 Along with the billing functions, the billing and management server 42 has a symmetric network management protocol ("SNMP") management console 126. The SNMP console 126 monitors failures in the systems and monitors systems resources.

Also shown is a backup billing and management server 128 and SNMP console 130. Should the first billing and management server 42 fail for any reason the backup continues to
15 operate. Thus, redundantly collecting billing and operations information.

Figure 6 shows another embodiment of the invention where a computer 140 accesses a LEC 12 and, through an Internet provider 142, connects to the Internet. As described thus far in this Figure, the connection to the Internet is well known in the art. Computer users often connect to the Internet by dialing from a modem in the computer 140 through a LEC 12 to an
20 Internet provider 142. The Internet provider 142 has a modem that communicates to the modem in the computer 140 and provides a gateway to the Internet 36 for the computer 140. The computer 140 can then access content servers 144 on the Internet 36 using a single IP stream.

In the invention, the callover units 40 are connectable by the computer 140 over the
25 Internet 36. The computer can then place phone calls through the appropriate callover unit 40 to a LEC 26 local to the callover unit 40 and then directly to a telephone 28. This is useful

under many circumstance. For example, the computer itself can now act as a telephone if for any reason the geographical area where the computer 140 is situated does not have a local callover unit. Further, the computer 140 can interact with a user on the telephone 28 to access information on one or more of the content servers 144, provided of course that the computer

5   140 accepts voice response. Another example is computer generated phone calling for marketing or dunning.

In the latter example, the content server 144 is initiating a phone call over the Internet 36 to a callover unit 40. Commercially available software allows the content server 144 to perform this task by using a modem to dial a phone number and transmits a message to the

10   phone 28. In order to operate though, most modems require a dial tone, ring indication, and other status messages from the telephone network. For this reason, the callover unit 40 artificially reproduces these status messages normally transmitted by the LEC 26.

For each of the other examples where the computer 140 initiates the call, the internet engine used by the computer 140 must be adapted to interface with the callover unit 40. The

15   adaptation is generally a software add-in which handles communication and billing by the callover unit 40.

If the user of the computer 140 does not have a subscription with an Internet provider the callover unit 32 can also be used to establish a connection to the Internet 36, as is shown in Figure 7. The computer 140 simply places a call using a an Internet access device such as a

20   device for communicating on ISDN, Ethernet, a modem, et cetera. The Internet access device then communicates through a LEC 12 to a callover unit 32. The callover unit 32 determines that the transmission is a data-type transmission which is not latent sensitive and provides a connection to the Internet 36. The computer 140 can then access content servers 144 on the Internet 36 as if the computer 140 were using an Internet provider, thus allowing

25   use of a system that charges per unit time as opposed to by subscription.

20

The software add-in in the preferred embodiment has a graphical user interface ("GUI") that is as shown in Figure 8. The software provides is a viewer that allows access to the Internet and to phone lines worldwide to every computer. This software also allows computers that do not have Internet service to communicate through Internet facilities to

5   those that do. The viewer can also interface to a debit card billing system or other third-party billing system from the callover units without limiting the third-party billing systems to those with Internet access.

The GUI shown is designed for use with WINDOWS, a trademark owned by Microsoft Corporation, as can be seen from the WINDOWS header 160. The GUI provides dialing

10   dialog box 162 which has a telephone-like keypad 166 for telephone number entry. As the numbers are keyed they are displayed in a calling window 164. The user then has various options presented by radio-buttons on the side of the keypad 166. The number can be dialed by clicking on the dial button 168. Dialing makes a connection through the Internet with a telephone as is illustrated in Figure 6, or to another computer as is shown in Figure 7.

15   When the call is complete, the hang-up button 170 can be used to disconnect the call and free resources for a subsequent caller.

The attach button is used to pre-configure other software applications to link to the data stream on execution. This allows third party software applications, such as a terminal emulator or a program manager, to run through the viewer thus establishing pier-to-pier data

20   communication. For example, an operator on one computer can then manipulate the software running on another computer remotely, using a remote control application linked over the Internet.

The buttons shown below the keypad 166 are for use once a call has been established. The hold button 176 temporarily pauses a call while the "conf" button 174 allows conference

25   call with multiple parties. The L1 and L2 buttons are used for telephone line one and

telephone line two for multi-party calls, though this assumes that the desktop has both sufficient bandwidth and/or sound card resources.

During the course of connection and conversation, various statuses are presented to the user. The status window 180 presents text messages to a user, the called number window 190

5   states the number that has been dialed, the call time window 192 lets the user know how long the call has been connected, and the call type window 194 presents a type of communication, such as fax, voice, data, et cetera. Connection status 196 also shows the current connection properties such as off the hook, busy or idle.

Audio levels can also be controlled using the audio level dialog box 182. This dialog

10 box allows a user to adjust audio volume in the computer's speaker system as well as the volume of audio transmitted.

If mail is to be sent over the Internet, a message composer dialog box is retrieved by pressing the message composer button 186. This allows access to a mail editor for sending written messages via the Internet.

15 Pressing the IVR script editor button 188 brings up a dialog box which allows the user to edit an IVR script. The IVR script allows a user to compose an IVR application. The user simply keys into a menu structure which keys on a keypad using DTMF corresponds to which incoming or outgoing messages. The outgoing messages can be recorded and stored in files if the computer has sound capabilities or can be typed and read using standard computer

20 diction. On of the many applications for this menuing system is to have a multi-user answering machine where each user only listens to messages directed toward him/her.

In the preferred embodiment a clock showing the time of day is also shown 184.

Figure 9 shows that the invention can also be used to provide interactive voice response ("IVR") to a user initiating a call on a standard telephone 10. The call passes through a

25 PSTN 120 to a callover unit 32. The call travels the Internet 36 as a packetized data to another callover unit 150.

within the meaning and range of equivalency of the claims are therefore intended to be embraced therein.

Claims

The embodiments of the invention in which an exclusive property or privilege is claimed are defined as follows:

1.   A telecommunication system for bi-directionally communicating information signals between a first communication device connected to a first telephone network and a second communication device, the telecommunication system comprising

     network means for communicating discrete packets of digital information to the second communication device; and

     first interface means in electrical communication with the first telephone network and the network means, the interface means having a transmitting mode for converting the information signals into the discrete packets of digital information and transmitting the discrete packets to the network means, and having a receiving mode for converting the discrete packets received from the network means into information signals and transmitting the information signals to the first communication device over the first telephone network.

2.   The telecommunication system according to claim 1 wherein the first telecommunications device is a computer having a modem adapted to communicate over the first telephone network.

3.   The telecommunication system according to claim 1 wherein the first telecommunications device is a telephone.

4.   The telecommunication system according to claim 1 wherein the second telecommunication device is a computer electrically connected to the network means.

5.   The telecommunication system according to claim 1 wherein the second communication device comprises second interface means in electrical communication with the network means and a second telephone network, the interface means having a transmitting mode for converting the information signals into the discrete packets of digital information and transmitting the discrete packets to the network means, and having a receiving mode for converting the discrete packets received from the

network means into information signals and transmitting the information signals over the second telephone network.

6. The telecommunication system according to claim 5 wherein the second communication device further comprises a telephone which communicates with the second interface means via the second telephone network.

7. The telecommunication system according to claim 5 wherein the second communication device further comprises a computer having a modem for communicating with the second interface means via the second telephone network.

8. The telecommunication system according to claim 5 wherein the second communication device is adapted to communicate status signals from the second telephone network over the network means to the first interface means.

9. The telecommunication system according to claim 1 wherein the interface means further comprises management means for determining a lowest average latency prior to transmission of the discrete packets.

10. The telecommunication system according to claim 9 wherein the management means transmits discrete packets from the information signal in a plurality of IP streams.

11. The telecommunication system according to claim 9 wherein the management means redundantly transmits each discrete packet from the information signal multiple times.

12. The telecommunication system according to claim 1 wherein the interface means further comprises compression means for compressing the information signal into a compressed signal having a smaller size than the information signal.

13. The telecommunication system according to claim 1 wherein the compression means compresses the information without loss of data.

14. The telecommunication system according to claim 1 wherein the interface means the further comprises voice response means for interactively determining a communication path for the information signal.

15.   The telecommunication system according to claim 14 wherein interface means transmits to the network means or to a long distance carrier in response to the voice response means.

16.   The telecommunication system according to claim 1 further comprising billing means electrically connected to the network means for generating time and billing information relating to usage of the interface means.

17.   The telecommunication system according to claim 16 wherein the interface means is distributed remotely from the billing means such that communication is performed over the network means.

18.   The telecommunication system according to claim 17 wherein the interface means temporarily stores the time and billing information and transmits the time and billing information to the billing means after communication between the first communication device and the second communication device terminates.

19.   The telecommunication system according to claim 1 wherein the interface means is adapted to interface with a third-party billing system.

20.   The telecommunication system according to claim 19 wherein the third-party billing system is adapted to electrically communicate over the network means.

21.   The telecommunication system according to claim 19 wherein the third-party billing system is not adapted to communicate over the network means.

22.   The telecommunication system according to claim 1 wherein the first communication device is a computer having a user interface which enables interaction of the computer over the network means with the second communication device.

23.   The telecommunication system according to claim 22 wherein the second communication device is a computer electrically connected to the network means.

24.   The telecommunication system according to claim 22 wherein the second communication device is a telephone and the telecommunication system further comprises:

second interface means in electrical communication with the network means, the
5   interface means having a transmitting mode for converting the information
signals into the discrete packets of digital information and transmitting the
discrete packets to the network means, and having a receiving mode for
converting the discrete packets received from the network means into
reassembled information signals; and

10   second local exchange means in electrical communication with the second
communication device for communicating the reassembled information signals
via a telephone line.

25.   The telecommunication system according to claim 1 wherein the first interface means
further comprises a database having a list of telephone exchanges serviced by the
second interface means.

26.   The telecommunication system according to claim 25 wherein the database further
comprises a list of IP addresses used by the second interface means.

27.   The telecommunication system according to claim 25 wherein the database further
comprises latency information for average latencies over segments of the network
means and the first interface means operating in the transmission mode is adapted to
avoid problematic segments as determined by the average latencies by using multiple
5   transmission methods for a single transmission.

28.   The telecommunication system according to claim 1 wherein the second
communication device is a content server in electrical communication with the
network means, the content server having a user interface that is adapted to interact
with the first communication device.

29.   The telecommunication system according to claim 28 wherein the first interface
means simulates dial tone to the content server.

30.   The telecommunication system according to claim 1 wherein the receiving mode
communicates over a first set of one or more IP streams and the transmit mode
communicates over a second set of one or more IP streams.

31. The telecommunication system according to claim 30 wherein the first set and the second set are mutually exclusive.

32. The telecommunication system according to claim 1 wherein the discrete packets are scaled to approximately match a maximum transfer rate of a transmission path in the network means.

33. The telecommunication system according to claim 1 wherein the discrete packets are formatted according to Internet Protocol.

34. The telecommunication system according to claim 1 wherein the network means is accessible with a subscription and the first interface means provides access to the network means without the subscription.

35. A telecommunications system for bi-directionally communicating between a first communication device connected to a first telephone network and a second communication device connected to a second telephone network over a computer network, the telecommunication system comprising

5   a first callover unit having a telephone connection to the first telephone network and having a network connection to the computer network such that electronic signals from the first telephone network are converted and transmitted on the computer network, and information packets from the computer network are converted and transmitted on the first telephone network; and

10   a second callover unit having a telephone connection to the second telephone network and having a network connection to the computer network such that electronic signals from the second telephone network are converted and transmitted on the computer network, and information packets from the computer network are converted and transmitted on the second telephone network, thereby providing

15   bi-directional communication between the first communication device and the second communication device over the computer network.

35. The telecommunication system according to claim 34 wherein the first callover unit further comprises voice response means for interactively determining a communication path for the information signal.

36. The telecommunication system according to claim 35 wherein first callover unit transmits to the second callover unit over the computer network or to a long distance carrier in response to the voice response means.

37. The telecommunication system according to claim 34 further comprising billing means electrically connected to the computer network for generating time and billing information relating to usage of the first callover unit.

38. The telecommunication system according to claim 37 wherein the first callover unit transmits the time and billing information to the billing means via the computer network.

39. A method of managing packet transmission a telecommunication system which transmits discrete packets of information, method comprising the steps of

   determining whether the packet transmission is latent sensitive; and

   minimizing latency if the packet transmission is latent sensitive.

40. The method according to claim 39 wherein the step of minimizing latency further comprises the step of determining an average latency of the packet transmission where test packets are sent over a plurality of IP streams and the and amount of transmission time is measured for each IP streams resulting in a fastest IP stream.

41. The method according to claim 40 wherein the step of minimizing latency further comprises the step of breaking the test packets into fractional pieces and distributing the fractional pieces over the plurality of IP streams.

42. The method according to claim 39 wherein the step of minimizing latency is performed in multiple passes where each pass transmits the discrete packets according to a predetermined method.

43. The method according to claim 42 wherein the predetermined method comprises a step of changing a size of the discrete packets.

44. The method according to claim 42 wherein the predetermined method comprises a step of changing a compression rate of the discrete packets.

45. The method according to claim 42 wherein the predetermined method comprises a step of changing a fractional amount of the information transmitted in the discrete packets.

46. The method according to claim 39 wherein the step of minimizing latency further comprises the step of transmitting the discrete packets over a first set of IP streams and receiving other discrete packets over a second set of IP streams.

47. The method according to claim 39 wherein the step of minimizing latency is repeated periodically throughout the packet transmission.

48. A telecommunications system for bi-directionally communicating between a first communication device connected to a first telephone network and a computer connected to a computer network, the telecommunication system comprising a callover unit having a telephone connection to the first telephone network and having a network connection to the computer network such that electronic signals from the first telephone network are converted and transmitted on the computer network, and information packets from the computer network are converted and transmitted on the first telephone network.

49. The telecommunications system of claim 48 wherein the callover unit communicates over the computer network using discrete packets of information.

50. The telecommunications system of claim 48 wherein the computer network is the Internet.

AMENDED CLAIMS
[received by the International Bureau on 23 June 1997 (23.06.97);
original claims 1 and 35 amended; remaining claims unchanged (8 pages)]

The embodiments of the invention in which an exclusive property or privilege is claimed
are defined as follows:

1.   A telecommunication system for bi-directionally communicating information signals
between a first communication device connected to a first telephone network and a
second communication device, the first communication device initiating a bi-
directional communication by providing an identifier of the second communication
device to the first telephone network as at least one of the information signals, the
telecommunication system comprising:

network means for communicating discrete packets of digital information to the
second communication device; and

first interface means in electrical communication with the first telephone network and
the network means, the interface means having a transmitting mode for
converting the information signals into the discrete packets of digital
information and transmitting the discrete packets to the network means, and
having a receiving mode for converting the discrete packets received from the
network means into information signals and transmitting the information
signals to the first communication device over the first telephone network, and
the interface means being adapted to receive the identifier from the first
telephone network and convert the at least one of the information signals
including the identifier such that the interface means directs the bi-directional
communication to the second communication device utilizing the identifier
received from the first telephone network when the first communication device
initiated the bi-directional communication.

2.   The telecommunication system according to claim 1 wherein the first
telecommunications device is a computer having a modem adapted to communicate
over the first telephone network.

3.   The telecommunication system according to claim 1 wherein the first
telecommunications device is a telephone.

32   AMENDED SHEET (ARTICLE 19)

4.   The telecommunication system according to claim 1 wherein the second telecommunication device is a computer electrically connected to the network means.

5.   The telecommunication system according to claim 1 wherein the second communication device comprises second interface means in electrical communication with the network means and a second telephone network, the interface means having a transmitting mode for converting the information signals into the discrete packets of digital information and transmitting the discrete packets to the network means, and having a receiving mode for converting the discrete packets received from the network means into information signals and transmitting the information signals over the second telephone network.

6.   The telecommunication system according to claim 5 wherein the second communication device further comprises a telephone which communicates with the second interface means via the second telephone network.

7.   The telecommunication system according to claim 5 wherein the second communication device further comprises a computer having a modem for communicating with the second interface means via the second telephone network.

8.   The telecommunication system according to claim 5 wherein the second communication device is adapted to communicate status signals from the second telephone network over the network means to the first interface means.

9.   The telecommunication system according to claim 1 wherein the interface means further comprises management means for determining a lowest average latency prior to transmission of the discrete packets.

10.   The telecommunication system according to claim 9 wherein the management means transmits discrete packets from the information signal in a plurality of IP streams.

11.   The telecommunication system according to claim 9 wherein the management means redundantly transmits each discrete packet from the information signal multiple times.

AMENDED SHEET (ARTICLE 19)

12.   The telecommunication system according to claim 1 wherein the interface means further comprises compression means for compressing the information signal into a compressed signal having a smaller size than the information signal.

13.   The telecommunication system according to claim 1 wherein the compression means compresses the information without loss of data.

14.   The telecommunication system according to claim 1 wherein the interface means the further comprises voice response means for interactively determining a communication path for the information signal.

15.   The telecommunication system according to claim 14 wherein interface means transmits to the network means or to a long distance carrier in response to the voice response means.

16.   The telecommunication system according to claim 1 further comprising billing means electrically connected to the network means for generating time and billing information relating to usage of the interface means.

17.   The telecommunication system according to claim 16 wherein the interface means is distributed remotely from the billing means such that communication is performed over the network means.

18.   The telecommunication system according to claim 17 wherein the interface means temporarily stores the time and billing information and transmits the time and billing information to the billing means after communication between the first communication device and the second communication device terminates.

19.   The telecommunication system according to claim 1 wherein the interface means is adapted to interface with a third-party billing system.

20.   The telecommunication system according to claim 19 wherein the third-party billing system is adapted to electrically communicate over the network means.

21.   The telecommunication system according to claim 19 wherein the third-party billing system is not adapted to communicate over the network means.

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AMENDED SHEET (ARTICLE 19)

22.   The telecommunication system according to claim 1 wherein the first communication device is a computer having a user interface which enables interaction of the computer over the network means with the second communication device.

23.   The telecommunication system according to claim 22 wherein the second communication device is a computer electrically connected to the network means.

24.   The telecommunication system according to claim 22 wherein the second communication device is a telephone and the telecommunication system further comprises:

5   second interface means in electrical communication with the network means, the interface means having a transmitting mode for converting the information signals into the discrete packets of digital information and transmitting the discrete packets to the network means, and having a receiving mode for converting the discrete packets received from the network means into reassembled information signals; and

10   second local exchange means in electrical communication with the second communication device for communicating the reassembled information signals via a telephone line.

25.   The telecommunication system according to claim 1 wherein the first interface means further comprises a database having a list of telephone exchanges serviced by the second interface means.

26.   The telecommunication system according to claim 25 wherein the database further comprises a list of IP addresses used by the second interface means.

27.   The telecommunication system according to claim 25 wherein the database further comprises latency information for average latencies over segments of the network means and the first interface means operating in the transmission mode is adapted to avoid problematic segments as determined by the average latencies by using multiple

5   transmission methods for a single transmission.

28.   The telecommunication system according to claim 1 wherein the second communication device is a content server in electrical communication with the network means, the content server having a user interface that is adapted to interact with the first communication device.

29.   The telecommunication system according to claim 28 wherein the first interface means simulates dial tone to the content server.

30.   The telecommunication system according to claim 1 wherein the receiving mode communicates over a first set of one or more IP streams and the transmit mode communicates over a second set of one or more IP streams.

31.   The telecommunication system according to claim 30 wherein the first set and the second set are mutually exclusive.

32.   The telecommunication system according to claim 1 wherein the discrete packets are scaled to approximately match a maximum transfer rate of a transmission path in the network means.

33.   The telecommunication system according to claim 1 wherein the discrete packets are formatted according to Internet Protocol.

34.   The telecommunication system according to claim 1 wherein the network means is accessible with a subscription and the first interface means provides access to the network means without the subscription.

35.   A telecommunications system for bi-directionally communicating between a first communication device connected to a first telephone network and a second communication device connected to a second telephone network over a computer network, the first communication device initiating a bi-directional communication by providing an identifier representative of the second communication device to the first telephone network transmitted as a part of electronic signals through the first telephone network, the telecommunication system comprising:

        a first callover unit having a telephone connection to the first telephone network having a network connection to the computer network such that electronic

10 signals from the first telephone network are converted and transmitted on the computer network, and information packets from the computer network are converted and transmitted on the first telephone network; and

a second callover unit having a telephone connection to the second telephone network and having a network connection to the computer network such that electronic

15 signals from the second telephone network are converted and transmitted on the computer network, and information packets from the computer network are converted and transmitted on the second telephone network, thereby providing bi-directional communication between the first communication device and the second communication device over the computer network, and the first

20 callover unit being adapted to receive the identifier from the first telephone network and selectively convert and transmit the part of the electronic signals including the identifier thus directing the bi-directional communication to the second communication device utilizing the identifier transmitted by the first communication device upon initiation of the bi-directional communication.

35. The telecommunication system according to claim 34 wherein the first callover unit further comprises voice response means for interactively determining a communication path for the information signal.

36. The telecommunication system according to claim 35 wherein first callover unit transmits to the second callover unit over the computer network or to a long distance carrier in response to the voice response means.

37. The telecommunication system according to claim 34 further comprising billing means electrically connected to the computer network for generating time and billing information relating to usage of the first callover unit.

38. The telecommunication system according to claim 37 wherein the first callover unit transmits the time and billing information to the billing means via the computer network.

AMENDED SHEET (ARTICLE 19)

39. A method of managing packet transmission a telecommunication system which transmits discrete packets of information, method comprising the steps of

determining whether the packet transmission is latent sensitive; and

minimizing latency if the packet transmission is latent sensitive.

40. The method according to claim 39 wherein the step of minimizing latency further comprises the step of determining an average latency of the packet transmission where test packets are sent over a plurality of IP streams and the and amount of transmission time is measured for each IP streams resulting in a fastest IP stream.

41. The method according to claim 40 wherein the step of minimizing latency further comprises the step of breaking the test packets into fractional pieces and distributing the fractional pieces over the plurality of IP streams.

42. The method according to claim 39 wherein the step of minimizing latency is performed in multiple passes where each pass transmits the discrete packets according to a predetermined method.

43. The method according to claim 42 wherein the predetermined method comprises a step of changing a size of the discrete packets.

44. The method according to claim 42 wherein the predetermined method comprises a step of changing a compression rate of the discrete packets.

45. The method according to claim 42 wherein the predetermined method comprises a step of changing a fractional amount of the information transmitted in the discrete packets.

46. The method according to claim 39 wherein the step of minimizing latency further comprises the step of transmitting the discrete packets over a first set of IP streams and receiving other discrete packets over a second set of IP streams.

47. The method according to claim 39 wherein the step of minimizing latency is repeated periodically throughout the packet transmission.

48. A telecommunications system for bi-directionally communicating between a first communication device connected to a first telephone network and a computer connected to a computer network, the telecommunication system comprising a callover unit having a telephone connection to the first telephone network and having

5   a network connection to the computer network such that electronic signals from the first telephone network are converted and transmitted on the computer network, and information packets from the computer network are converted and transmitted on the first telephone network.

49. The telecommunications system of claim 48 wherein the callover unit communicates over the computer network using discrete packets of information.

50. The telecommunications system of claim 48 wherein the computer network is the Internet.
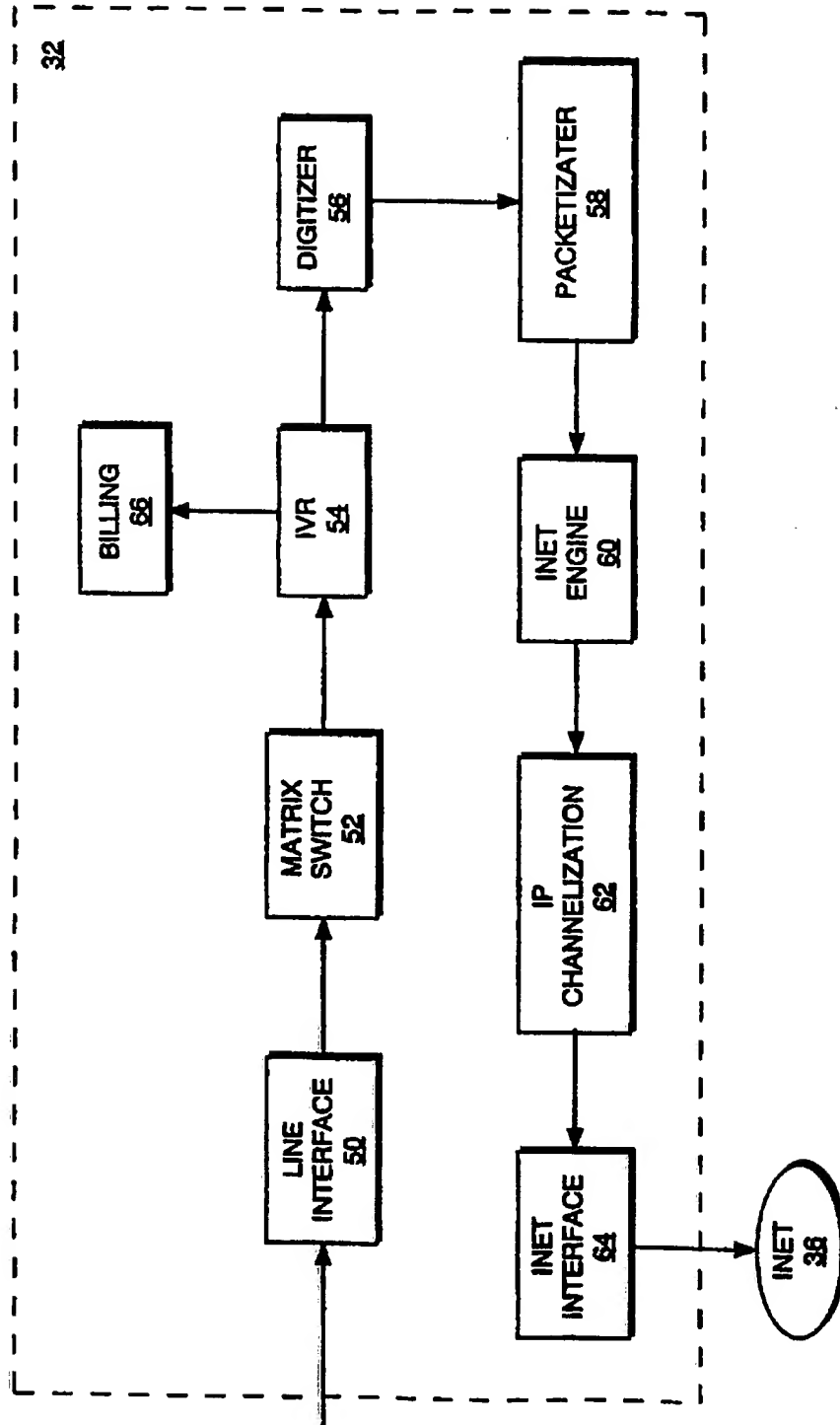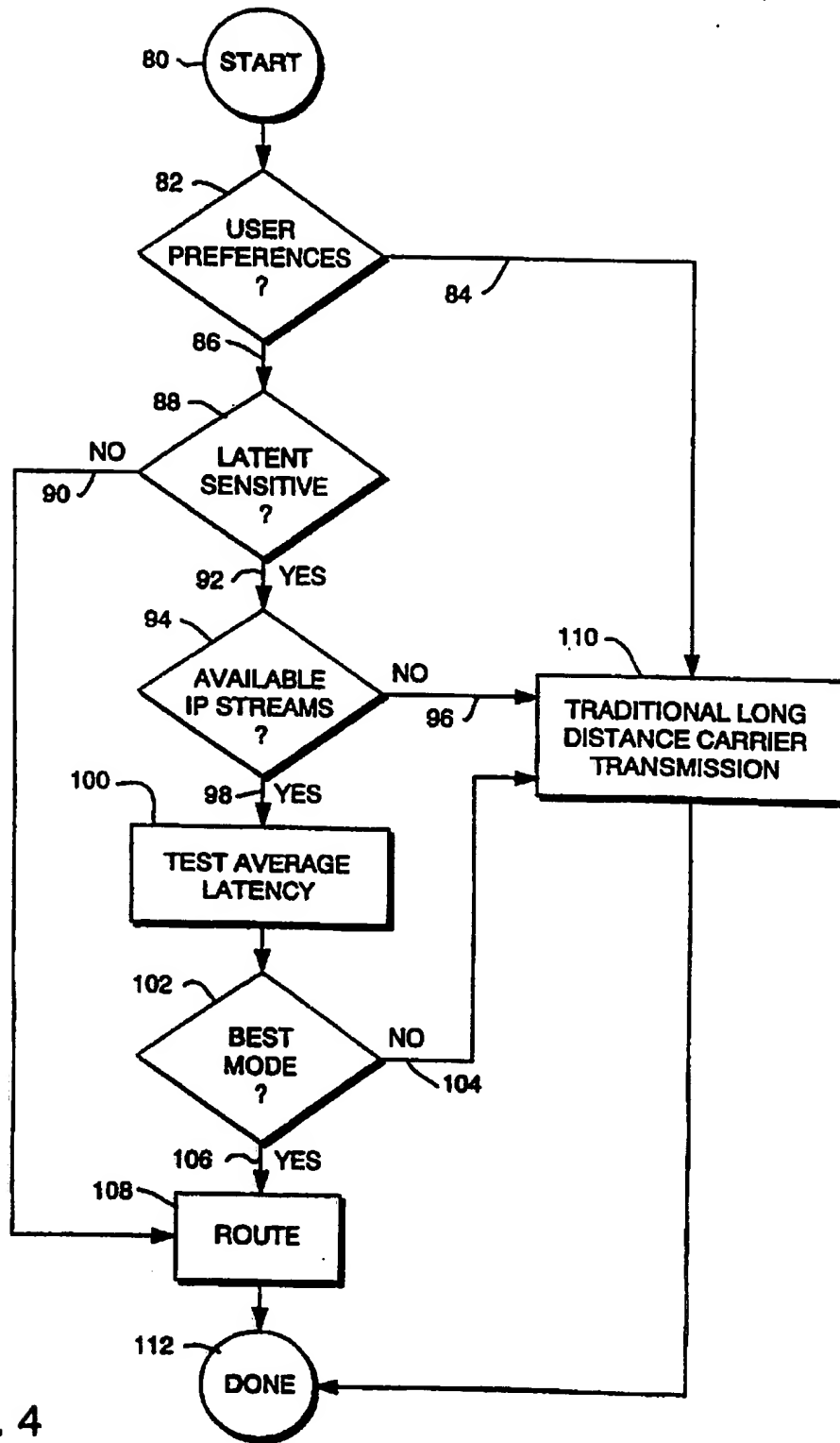
1/7
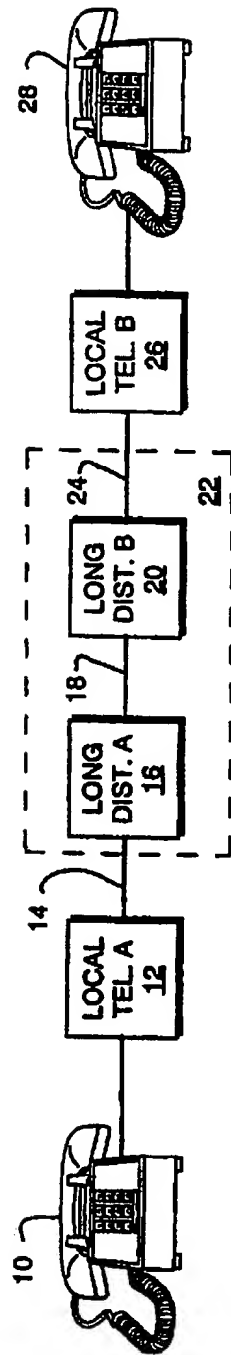
FIG. 1
(PRIOR ART)
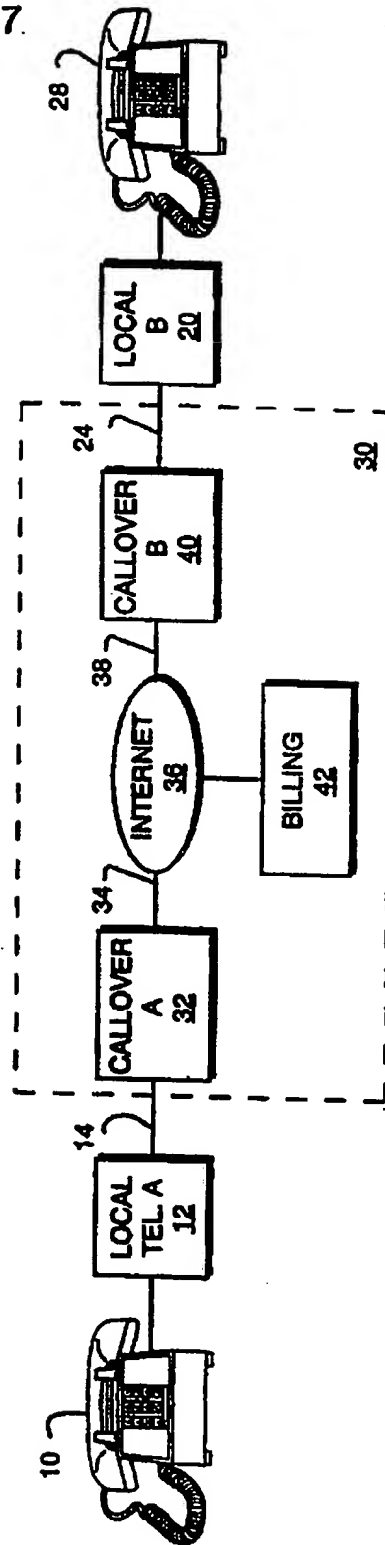
FIG. 2

FIG. 3

FIG. 4

4 / 7
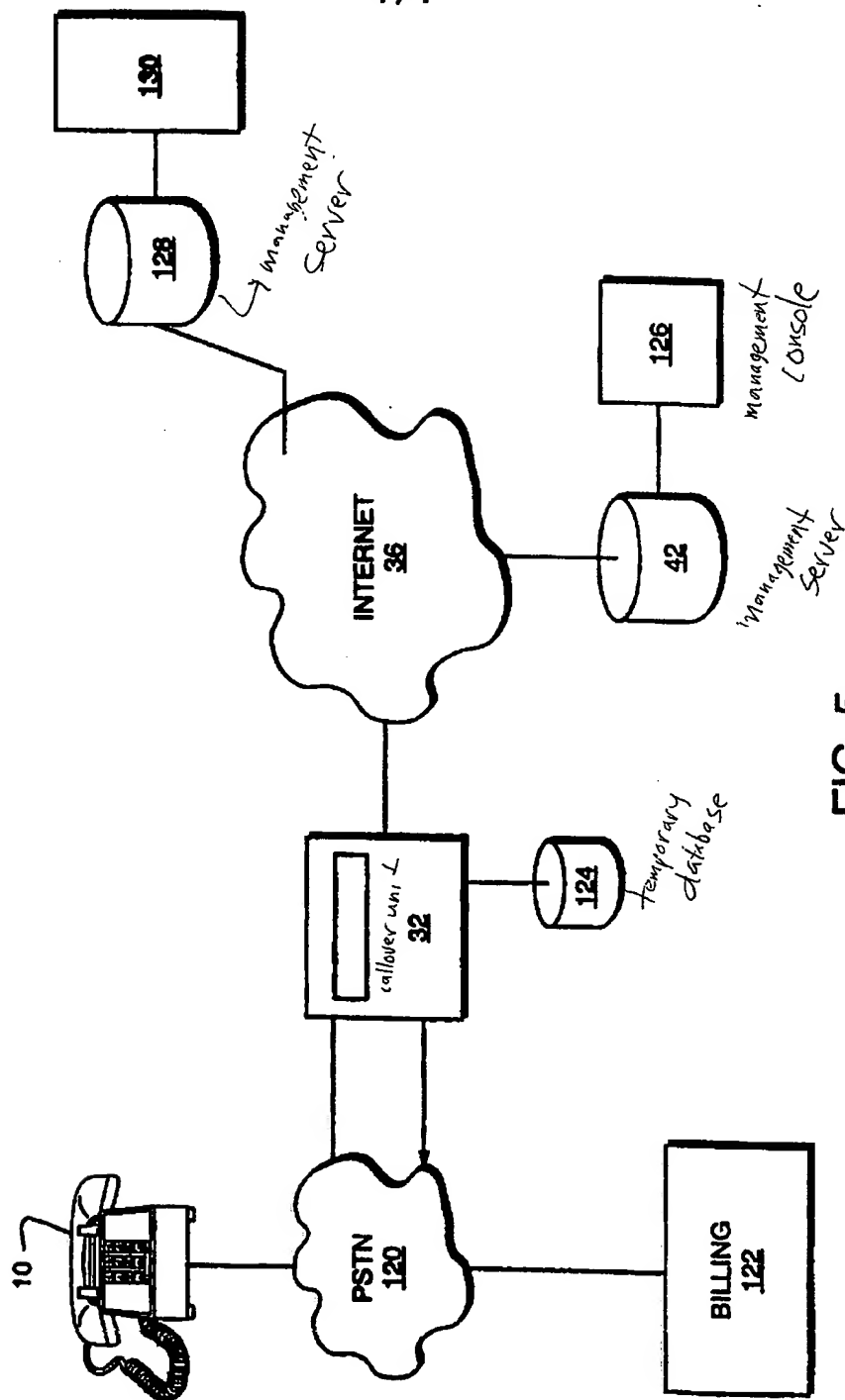
FIG. 5
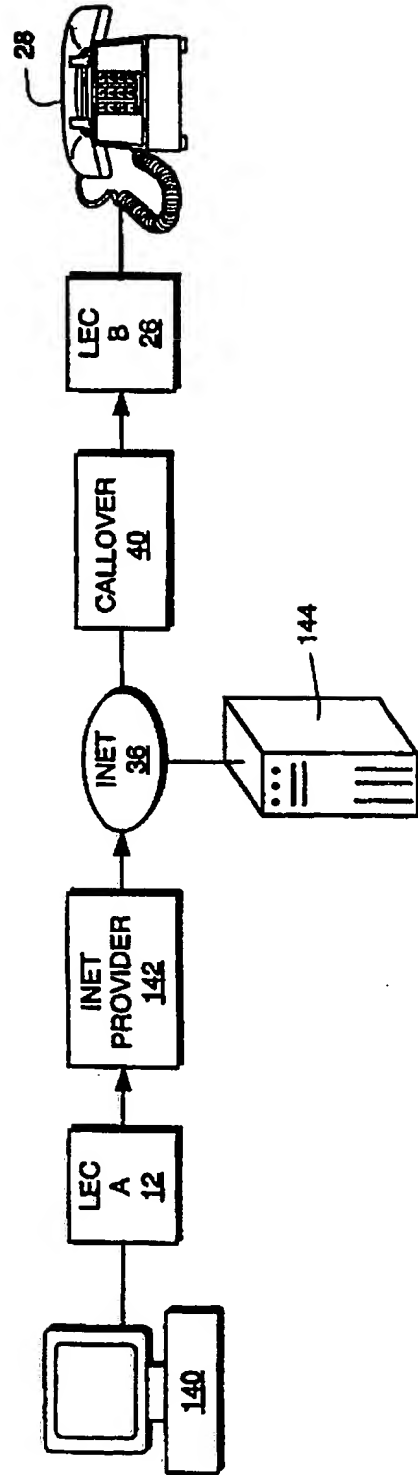
FIG. 6
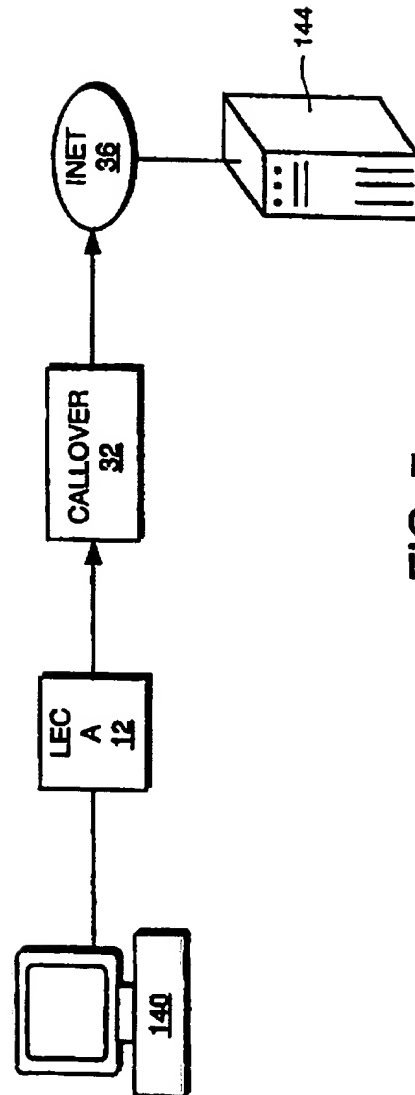
FIG. 7
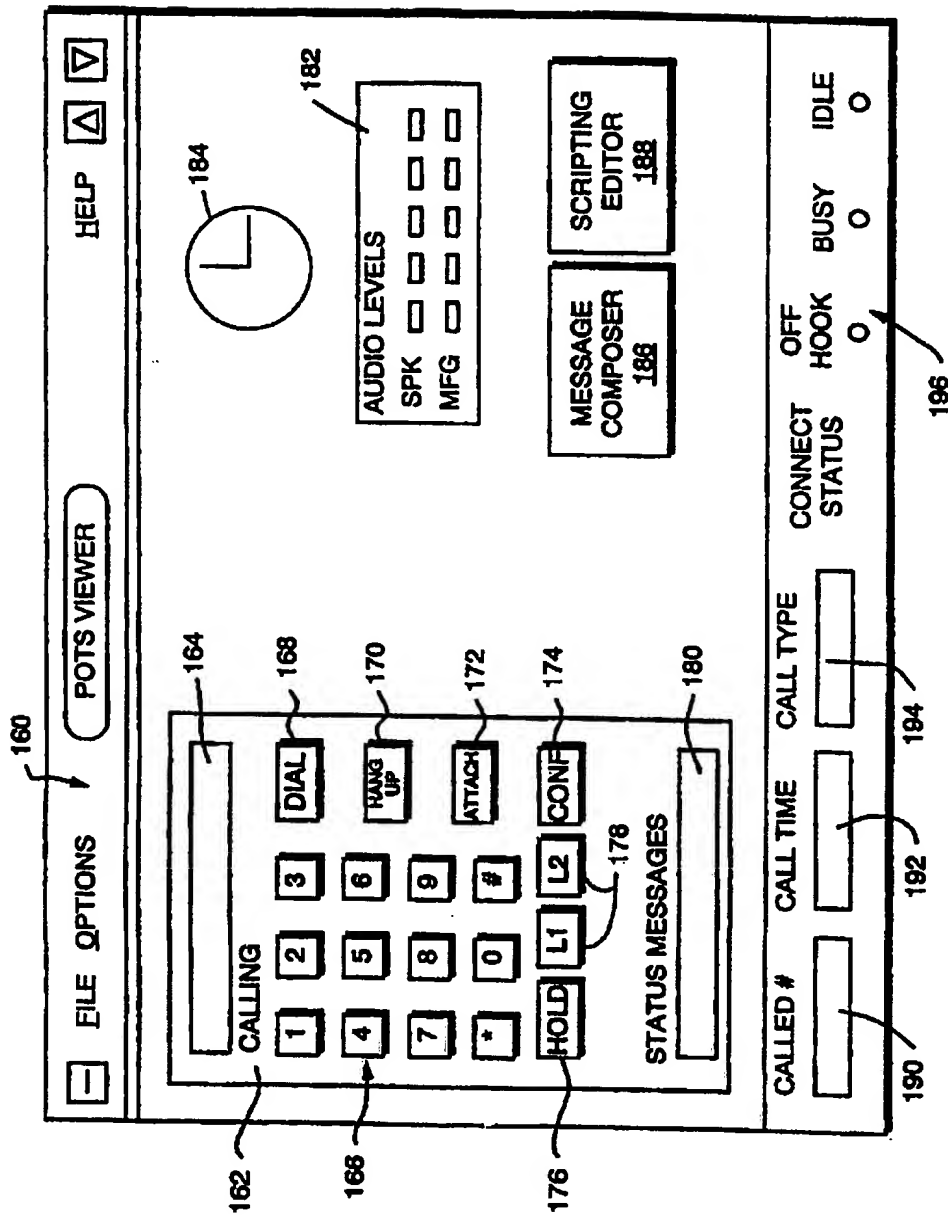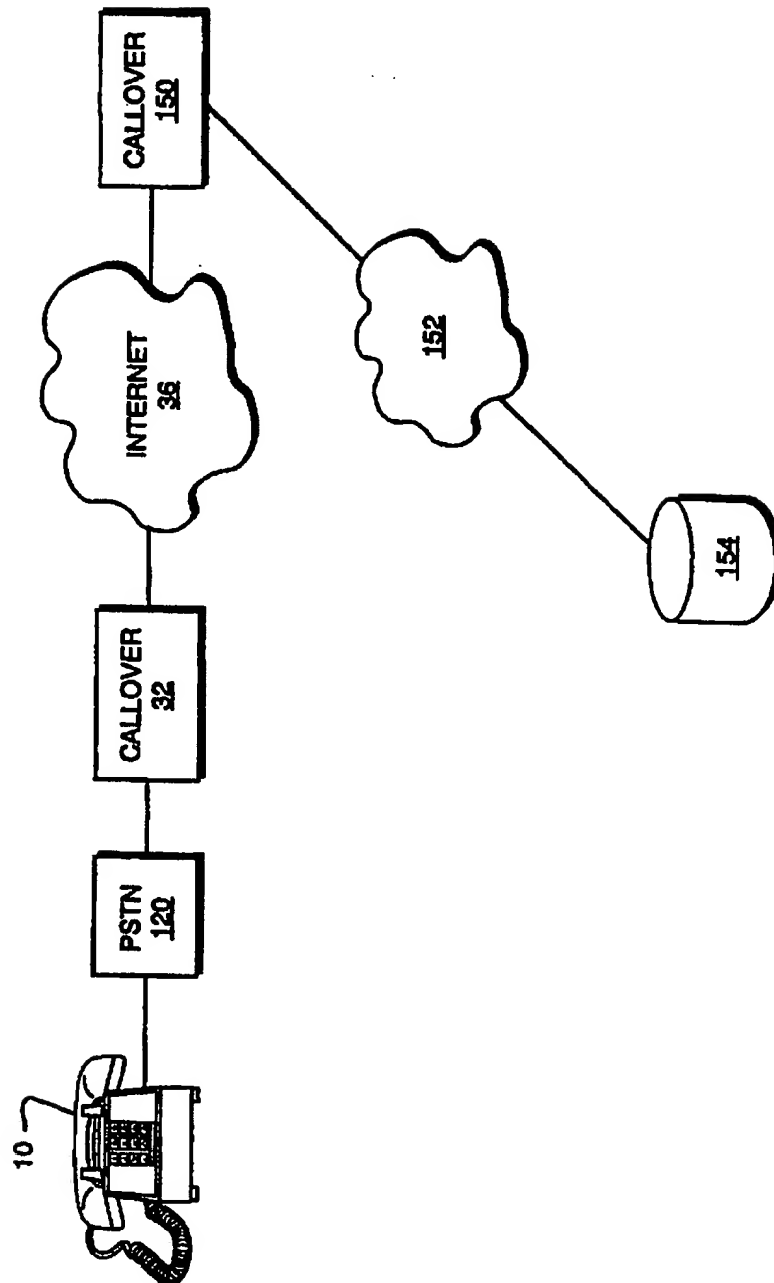
FIG. 8

FIG. 9

INTERNATIONAL SEARCH REPORT

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6)   :H04L 12/56
US CL   :370/389

According to International Patent Classification (IPC) or to both national classification and IPC

B.   FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. :   370/345, 351, 389, 392, 402, 465, 466

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

NONE

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS (Internet, packet network, callover, callback, call back, telephone, Internet telephone, Internet phone)

C.   DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	YANG, C. INETPhone: Telephone Services and Servers on Internet, Network Working Group, RFP 1789. April 1995, pgs. 1-6.	1-50
A, E	US 5,608,786 A (GORDON) 04 March 1997, col. 8, line 62 to col. 9, line 17.	1-50
X	US 4,969,184 A (GORDON ET AL) 06 November 1990, col. 1, lines 38-56.	1, 35
A, P	US 5,526,353 A (HENLEY ET AL) 11 June 1996, col. 5, lines 1-30.	1, 35
A	US 5,353,283 A (TSUCHIYA) 04 October 1994.	1, 35
A	US 4,903,261 A (BARAN ET AL) 20 February 1990.	1, 35

☐ Further documents are listed in the continuation of Box C.   ☐ See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be of particular relevance		
"E"	earlier document published on or after the international filing date	"X"	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)		
"O"	document referring to an oral disclosure, use, exhibition or other means	"Y"	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 APRIL 1997	2 2 APR 1997

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	WELLINGTON CHIN
Facsimile No. (703) 305-3230	Telephone No. (703) 305-4366

Form PCT/ISA/210 (second sheet)(July 1992)